IN THE CLAIMS

1. (Currently Amended) A method of signal processing signals having transmission path characteristics, comprising the steps of:

inverse filtering an input signal having transmission path characteristics before processing the input signal wherein the transmission path characteristics of the input signal are reduced; and

processing the input signal;

wherein an inverse filter is used to filter the input signal and an encoder is used to process the input signal, the inverse filter being in communication with the encoder;

the inverse filter having an inverse amplitude response of a filter described by $h(t)$, the filter approximating noisy ambient conditions including telephone-channel-bandwidth conditions and the pre-filter response being characterized by:

$G(\omega) \approx \frac{1}{|H(\omega)|}$ wherein $H(\omega)$ is the frequency response of $h(t)$ and $G(\omega)$ is the inverse filter frequency response.

2. (Canceled).

3. (Currently Amended) The method of claim ~~2~~ 1, wherein the encoder is a Multi-Band Excitation (MBE) encoder and the inverse filter produces a frequency response having a smooth middle portion with peakiness at extremities of the frequency response.

4. (Currently Amended) The method of claim ~~2~~ 1, wherein the inverse filter comprises an all pole filter.

5. (Original) The method of claim 4, wherein the inverse filter is a low order filter including about six poles.

6. (Canceled)

7. (Currently Amended) The method of claim ~~6~~ 1, wherein a random signal having a power density characterized by $|G(\omega)|^2$ is used to design the inverse filter, wherein the processing step comprises the sub-steps of:

parameterizing the input signal; and

encoding the input signal; and the processing the signal method further comprises the steps of:

preprocessing the encoded signal; and

decoding the preprocessed encoded signal, wherein a parameter preprocessor is used to preprocess the encoded signal and a decoder is used to decode the preprocessed encoded signal, the encoder being in communication with the parameter preprocessor and the parameter preprocessor being in communication with the decoder.

8. (Currently Amended) A method for preprocessing a signal having transmission path characteristics, comprising the steps of:

obtaining ~~at least one~~ a first sequence, wherein one of the at least one obtained sequence is a first sequence [$h(n)$] wherein n = 0,1, . . . N-1, and N-1 is a length value of the first sequence;

obtaining a second sequence [$h_1(n)$] that modifies the first sequence [$h(n)$], the second sequence having a length M and the M length value being equal to a closest power of 2 after the N-1 length value;

wherein the FFT is taken on the second obtained sequence [$h_1(n)$] to determine $H(k)$

taking a Fast Fourier Transform (FFT) of ~~an~~ the second obtained sequence to determine $H(k)$;

obtaining $P(k)$ by using $H(k)$, wherein $P(k)$ is characterized by:

$$P(k) = \frac{1}{|H(k)|^2}$$, k=0,1,…,M-1~~, wherein M is a length value~~;

taking an inverse Fast Fourier Transform (IFFT) of $P(k)$ to obtain $R(m)$, wherein m = 0, 1, … M-1;

preparing Yule-Walker equations using the obtained $R(m)$ values;

solving the Yule-Walker equations to obtain coefficients;

using the obtained coefficients to design an inverse filter; and

preprocessing the signal having transmission path characteristics with the inverse filter.

9. (Original) The method of claim 8 wherein the using the obtained coefficients to design the inverse filter comprises the sub-steps of:

using the obtained coefficients to determine $G(\omega)$, wherein $G(\omega)$ is a frequency response of the inverse filter, and wherein $G(\omega) \approx \frac{1}{|H(\omega)|}$, $H(\omega)$ being the frequency response of $h(t)$, $h(t)$ being a time domain description of a filter that approximates transmission path characteristics including telephone-channel-bandwidth conditions, and $h(n)$ being a sequence representing the approximating filter;

using $G(\omega)$ to determine $g(t)$, wherein $g(t)$ is the time domain description of the inverse filter; and

using $g(t)$ to design the inverse filter.

10. (Original) The method of claim 9 wherein $|G(\omega)|^2$ is characterized by the equation

$$|G(\omega)|^2 = \frac{1}{|1 + \sum_{k=1}^{p} a_k e^{-j\omega k}|^2}$$

wherein a_k are the obtained coefficients.

11. (Canceled)

12. (Original) The method of claim 8 wherein the Yule-Walker equations are characterized by:

$$\begin{bmatrix} R(0) & R(-1) & \cdots & R(-p) \\ R(1) & R(0) & & \vdots \\ \vdots & & \ddots & \\ R(p) & R(p-1) & \cdots & R(0) \end{bmatrix} \cdot \begin{bmatrix} 1 \\ a_1 \\ a_2 \\ \vdots \\ a_p \end{bmatrix} = \begin{bmatrix} \sigma_p^2 \\ 0 \\ 0 \\ \vdots \\ 0 \end{bmatrix}$$

wherein σ_p^2 is a minimum mean-squared error of an auto recursive model, and $a_1, \ldots, a_p$ are the coefficients to be solved for.

13. (Original) The method of claim 8 wherein the inverse Fast Fourier Transform (IFFT) of $P(k)$ is characterized by:

$$\mathrm{IFFT}(P(k)) = R(m) = \sum_{k=0}^{K} \frac{1}{|H(k)|^2} * e^{\frac{j2\pi km}{K}}.$$

14. (Original) The method of claim 8 wherein the Yule Walker equations are solved by assuming a 'q' order Auto Recursive model and using a Levinson-Durbin method.

15. (Original) A method of preprocessing a signal having transmission path characteristics, comprising the steps of:

obtaining a frequency response $[H(\omega)]$ of a filter that approximates noisy ambient conditions including telephone-channel-bandwidth conditions;

modeling $|H(\omega)|^2$ using a moving average model comprising the sub-steps of:

taking the inverse Fast Fourier Transform (IFFT) of $|H(\omega)|^2$ to formulate a set of equations;

solving the set of equations to obtain moving average model parameters;

using the moving average model parameters to design an inverse filter; and

preprocessing the signal having transmission path characteristics with the inverse filter.

16. (Original) The method of claim 15 wherein the using the moving average model average model parameters to design the inverse filter comprises the sub-steps of:

applying the parameters to the equation:

$$|G(\omega)|^2 = \frac{1}{|1+\sum_{k=1}^{p} a_k e^{-j\omega k}|^2} = \frac{1}{|H(\omega)|^2}$$

wherein $G(\omega)$ is the frequency response of the pre-filter and a_k are the model parameters; and

using $G(\omega)$ to design the inverse filter.

17. (Currently Amended) A method of processing received encoded data, comprising the steps of:

preprocessoring the received encoded data before decoding the data, wherein the preprocessoring the received encoded data step includes the sub-steps of:

obtaining signal data from the received encoded data, wherein the obtained data includes pitch parameter data for a trajectory of successive frames of the signal;

removing at least one pitch parameter departure from the trajectory of successive frames;

smoothing the trajectory;

calculating at least one multiple corresponding to an obtained pitch parameter of a frame having a pitch parameter departure and at least one sub-multiple corresponding to the obtained pitch parameter;

comparing a pitch parameter from the removed and smoothened trajectory that corresponds to the obtained pitch parameter with the at least one corresponding multiple and the at least one corresponding sub-multiple; and

replacing the obtained pitch parameter with a new pitch parameter based on the comparison, the new pitch parameter being selected from the at least one corresponding multiple and the at least one corresponding sub-multiple; and

decoding the data.

18. (Original) The method of claim 17 wherein a parameter preprocessor performs the preprocessoring step and a decoder performs the decoding step, the parameter preprocessor being in communication with the decoder.

19. (Original) The method of claim 18 wherein the received encoded data is received from an encoder, the encoder being in communication with the parameter preprocessor.

20. (Canceled)

21. (Currently Amended) The method of claim ~~20~~ 17, wherein two multiples are calculated for each obtained pitch parameter of a frame having a pitch departure, two sub-multiples are calculated for each obtained pitch parameter of a frame having a pitch departure, the comparing step is performed for each obtained pitch parameter of a frame having a pitch departure, and the replacing step is performed for each obtained pitch parameter of a frame having a pitch departure.

22. (Currently Amended) The method of claim ~~20~~ 17, wherein a ~~medium~~ median filter is used to remove departures and a small order linear filter is used to smoothen.

23. (Currently Amended) The method of claim 22, wherein the obtained data further includes unvoiced frame and voiced frame information, and wherein the ~~medium~~ median filter and the small order linear filter are turned off when three successive unvoiced frames are detected.

24. (Currently Amended) The method of claim ~~20~~ 17, wherein the obtained data further includes spectral amplitude information; and wherein the preprocessoring the received encoded data step further includes the sub-steps of:

adjusting a number of harmonics for a spectrum of a frame having a new pitch parameter.

25. (Original) The method of claim 24, wherein the adjusting a number of harmonics step includes the sub-steps of:

removing each (2k-1)th harmonic of the spectrum if the new pitch parameter is one-half the value of the obtained pitch parameter;

removing each (3k-1)th harmonic and each (3k-2)th harmonic of the spectrum if the new pitch parameter is one-third the value of the obtained pitch parameter;

inserting one harmonic at each (k + 1/2) location of the spectrum if the new pitch parameter is twice the value of the obtained pitch parameter, each inserted (k+1/2)th harmonic having an amplitude characterized by the equation A(k+1/2) = $\sqrt{A(k) * A(k+1)}$; and

inserting one harmonic at each (k+1/3) and one harmonic at each (k+2/3) location of the spectrum if the new pitch parameter is three times the value of the obtained pitch parameter, each inserted (k+1/3)th harmonic having an amplitude characterized by the equation A(k + 1/3) = $\sqrt[3]{A^2(k)A(k+1)}$ and each inserted (k+2/3)th harmonic having an amplitude characterized by the equation $A(k+2/3)=\sqrt[3]{A(k)A^2(k+1)}$.

26. (Currently Amended) The method of claim 24, wherein the obtained data further includes voice parameter information; and

wherein the preprocessoring the received data step further includes the sub-steps of:

~~medium~~ median filtering a voice parameter trajectory, the voice parameter trajectory including voice parameter information of the frame having a new pitch parameter, voice parameter information of frames preceding the frame having a new pitch parameter, and voice parameter information of frames succeeding the frame having a new pitch parameter;

linear filtering the voice parameter trajectory;

using the ~~medium~~ median and linear filtered voice parameter trajectory to obtain a new voice parameter trajectory.

Claims 27-29 (Canceled).

30. (Currently Amended) A speech system comprising:

an inverse filtering means for inverse filtering signal data having transmission path characteristics;

an encoder, the encoder including parameterizing means for parameterizing the signal data and encoding means for encoding the signal data, the encoder being in communication with the inverse filtering means;

a parameter preprocessor, the parameter preprocessor including receiving means for receiving the encoded signal data and preprocessoring means for preprocessoring the received encoded signal data, the preprocessoring means including:

means for obtaining signal data from the received encoded data, wherein the obtained data includes pitch parameter data for a trajectory of successive frames of the signal;

means for removing at least one pitch parameter departure from the trajectory of successive frames;

means for smoothing the trajectory;

means for calculating at least one multiple corresponding to an obtained pitch parameter of a frame having a pitch parameter departure and at least one sub-multiple corresponding to the obtained pitch parameter;

means for comparing a pitch parameter from the removed and smoothened trajectory that corresponds to the obtained pitch parameter with the at least one corresponding multiple and the at least one corresponding sub-multiple; and

means for replacing the obtained pitch parameter with a new pitch parameter based on the comparison, the new pitch parameter being selected from the at least one corresponding multiple and the at least one corresponding sub-multiple

the parameter preprocessor being in communication with the encoder;

a decoder, the decoder including decoding means for decoding the preprocessed signal data and synthesizing means for synthesizing the preprocessed signal data into a speech signal, the decoder being in communication with the parameter preprocessor.